ABSTRACT

The present invention relates to methods for improving speech quality in e.g. an IP-telephony system. The invention reduces audio artefacts being due to overrun or underrun in a playout buffer caused by the sampling rates at a sending and receiving side not being at the same rate. The inventive solution modifies an LPC-residual on a sample-by-sample basis. The LPC-residual block comprising N samples is converted to a block comprising N+1 or N-1 samples. A sample rate controller 400 decides whether samples should be added to or removed from the LPC-residual. The exact position where to add respective remove samples is either chosen arbitrarily or found by searching for low energy segments in the LPC-residual. A speech synthesiser module 430 then reproduces the speech. By using the proposed sample rate conversion method the playout buffer 440 can be continuously controlled. Furthermore, since the method works on a sample-by-sample basis the buffer can be kept to a minimum and hence no extra delay is introduced.

(Publication figure: Figure 4)

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